

# Cisco 300-815 Exam | Latest 300-815 Exam Preparation - Money Back Guaranteed of Pass4sure 300-815 Study Materials



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We learned that a majority of the candidates for the 300-815 exam are office workers or students who are occupied with a lot of things, and do not have plenty of time to prepare for the 300-815 exam. Taking this into consideration, we have tried to improve the quality of our 300-815 training materials for all our worth. Now, I am proud to tell you that our 300-815 Training Materials are definitely the best choice for those who have been yearning for success but without enough time to put into it. There are only key points in our 300-815 training materials.

Cisco 300-815 (Implementing Cisco Advanced Call Control and Mobility Services) Certification Exam is a professional-level certification exam that is designed to test the knowledge and skills of IT professionals who are involved in implementing and managing advanced call control and mobility services using Cisco technologies. 300-815 Exam is one of the six concentration exams that you can choose from when pursuing the Cisco Certified DevNet Professional Certification.

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## Pass4sure 300-815 Study Materials - New 300-815 Exam Guide

In order to ensure the quality of 300-815 actual exam, we have made a lot of efforts. Our company spent a great deal of money on hiring hundreds of experts and they formed a team to write the work. The qualifications of these experts are very high. They have rich knowledge and rich experience on 300-815 study guide. These experts spent a lot of time before the 300-815 Study Materials officially met with everyone. And we have made scientific arrangements for the content of the 300-815 actual exam. You will be able to pass the 300-815 exam with our excellent 300-815 exam questions.

Cisco 300-815 Exam covers a wide range of topics, including call control protocols, Cisco Unified Communications Manager, Cisco Unity Connection, Cisco Unified IM and Presence, Cisco Expressway, and more. It is designed to test candidates' ability to implement and troubleshoot these technologies in a real-world environment, as well as their ability to design and deploy advanced call control and mobility services.

## Cisco Implementing Cisco Advanced Call Control and Mobility Services Sample Questions (Q45-Q50):

### NEW QUESTION # 45

An engineer must implement call restriction to toll-free numbers using a class of restriction in a branch Cisco UCME. In which two places is the corlist incoming or cor Incoming command configured? (Choose two.)

- A. "voice register pool" configuration mode

- B. "dial-peer cor custom" configuration mode
- C. "ephone-dn" configuration mode
- D. "voice register global" configuration mode
- E. "telephony-service" configuration mode

**Answer: A,C**

#### NEW QUESTION # 46

An engineer is configuring a Cisco Collaboration system for SIP endpoints and must enable Survivable Remote Site Telephony for these endpoints. Which code completes this configuration on the SRST gateway?

- A. voice register global  
default mode  
no allow-hash-in-dn  
max-dn 100  
max-pool 200
- B. telephony-service  
max-conferences 8 gain -6  
ip source-address 10.10.10.100 port 2000  
max-ephones 100  
max-dn 200
- C. voice service voip  
default mode secure  
address hiding  
allow-connections sip to sip  
sip registrar
- D. call-manager-fallback  
max-conferences 8 gain -6  
ip source-address 10.10.10.100 port 2000  
max-ephones 100  
max-dn 200

**Answer: D**

#### NEW QUESTION # 47

Refer to the exhibit. An administrator is trying to test outbound calls toward the ITSP but cannot complete the call and receives a SIP error. ITSP is consulted, and the issue is that the ANI that is being sent is not the DID provided 8005532447. Which configuration change sends the correct ANI on the INVITE sent to ITSP to fix the error?

- A. voice class sip-profiles 2  
request INVITE sip-header To modify "sip:(.\*)@" sip:8005532447@
- B. voice class sip-profiles 1  
request INVITE sip-header Diversion modify "sip:(.\*)@" sip:8005532447@"
- C. voice translation-rule 4  
rule 1/

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